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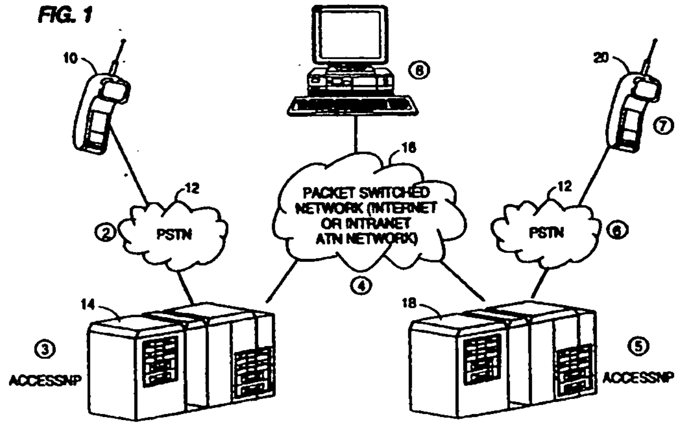
(54) A packet-switched-network telephone system

(57) A telephone system is disclosed establishing a link over a packet-switched-network (16), such as the Internet, between two telephone interfaces (10,20) by exchanging network addresses. The addresses can include ATM addresses to allow a virtual connection to be established between the interfaces. The interfaces exchange speech data packets of speech signals from a called party telephone and a calling party telephone over the network. An interface converts the speech signals of the parties into the speech data packets. The system dynamically allocates network and interface resources to complete the telephone call transaction. The system provides a number of enhanced calling services not normally available to PSTN subscribers including: message-delivery, call-back, call-screening, network-camp-on, call-back-from-queue, find-me, meet-me-pager, call-forwarding, call-interrupt, call-waiting, business-dialing, automatic-call-distribution, virtual-automatic-call-distribution, speed-dial, call-transfer, conference-calling and network-administration. Echo cancellation is used to eliminate echoes caused by PSTN circuits associated with the telephones. Silence detection is used to improve network efficiency by not sending packets when the parties are not speaking. The delay in the packet-switched-network is used to adjust the size of sound segments formed into a packet. The interfaces allow the system to provide end points in a packet-switched network that manage a telephone call over the packet switched network.

FIG. 1



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without requiring the called or calling parties to have special equipment.

[0011] The present invention typically provides telephone to telephone based, real-time, packet switched conversation.

[0012] The present invention typically improves voice channel transmission efficiency by not transmitting soundless portions of a telephone call.

[0013] The present invention typically uses connection protocols to reserve bandwidth or allocate network resources for the call so that delay is minimized in the packet-switched-network.

[0014] The present invention typically provides advanced telephone services, such as call forwarding or call queuing, for any telephone.

[0015] The present invention typically provides a smart, computationally powerful telephone system that can provide features that traditional systems, such as PBXs, cannot provide, such as screen based administration used for setting up an exclusion list to reject calls from selected calling parties.

[0016] The present invention typically provides enhanced calling services to take advantage of an inexpensive digital network for sophisticated communication between the calling and called parties.

[0017] The present invention typically performs compression and decompression of the voice signal transmitted over the packet-switched-network to reduce network usage.

[0018] The present invention typically suppresses echo signals occurring over the network when analog telephone circuits having hybrid circuits are part of the communication path.

[0019] The present invention typically provides a system that works with hardware commonly available and deployed today.

[0020] The present invention typically provides a system that does not require the subscriber to obtain special equipment, such as a computer or a simultaneous voice and data modem.

[0021] The present invention typically adjusts packet size of speech packets according to network delay.

[0022] The present invention typically provides end points in a packet-switched network that manage a telephone call over the packet switched network while dynamically allocating network resources as needed to complete the call transaction responsive to the service being provided.

[0023] The present invention typically provides a system that automatically places toll calls over the packet-switched network.

[0024] These together with other advantages which will be subsequently apparent, reside in the details of construction and operation as more fully hereinafter described and claimed, reference being had to the accompanying drawings forming a part hereof, wherein like numerals refer to like parts throughout.

Figure 1 illustrates the system components in an digital network telephone call;

Figure 2 depicts the components of the system of the present invention;

Figure 3 depicts another embodiment for handling toll calls;

Figure 4 depicts the operations that occur on a source side (platform 14) during a telephone call;

Figure 5 depicts the operations that occur on a destination side (platform 18) during a telephone call;

Figure 6 depicts the operations of the Message-Delivery enhanced service;

Figure 7 depicts a Camp-On service;

Figure 8 depicts a Find-Me service;

Figure 9 depicts the system in an intracompany environment; and

Figure 10 illustrates echo cancellation.

[0025] In a typical telephone call flow, in accordance with an embodiment of the present invention, a calling party subscriber using conventional telephone 10, as illustrated in Figure 1, calls through a public switched telephone network (PSTN) 12 to an originating or source service platform 14. The source platform 14 answers the call with an application that asks the caller to identify himself (account number, pass code, etc.), or uses automatic number identification (ANI) from the PSTN 12 to identify the caller/calling party telephone. If necessary, the source platform 14 asks the caller for the telephone number of the person they are trying to reach. The source platform 14 looks up the telephone number in a table to determine what equipment is servicing the area for the telephone number input by the caller. Using a packet-switched-network 16, such as the Internet that connects a system or group of platforms, the source platform 14 communicates with a destination platform 18 servicing the area of the called telephone. This communication involves identifying the called and calling parties, identifying the originating source platform 14, and determining real-time delay characteristics of the network connecting the two platforms 14 and 18. The two end points 14 and 18 manage the telephone call over the network 16.

Once the destination platform 18 has all of the appropriate information, it places an outgoing call over the PSTN 12 to the number or called party the caller asked to be dialed. The destination platform 18 monitors the outgoing call for ringing/busy/answer condition. Prior to the call being answered, the destination platform 18 indicates call status to the originatng source platform 14 and the source platform 14 can indicate the status to the caller such as by playing a ringing signal. When the call is answered via a telephone 20, the destination platform 18 indicates this to the source platform 14, the source platform 14 stops playing ringing tones to the caller and both systems begin the process of compressing the incoming speech from their respective PSTN connections, shipping it via packet-switched data packets to the

the called party answered the phone. Rather than accept the call, the called party can choose to direct the caller to the called party's voice mailbox, which is resident on the destination platform 18, to leave a message. Call screening dynamically allocates the resources of the destination platform responsive to the service being initiated. The destination, or the side of initiator of the service (in this case the destination where the subscriber has subscribed to call screening), manages the service.

[0035] Audible call screening can also be performed where the called party is allowed to listen to a message being recorded as is discussed in the disclosure entitled A Telephone System Integrating ... And Call Answering System previously mentioned.

[0036] The Find-Me service is a feature where the called party is a subscriber and has several telephone numbers where the subscriber may be reached (home, office, car). The destination platform 18 attempts each of the numbers in succession, or attempts all of the numbers at the same time, in the hopes of contacting the subscriber.

[0037] Network-Camp-On is a feature by which the destination platform 18 monitors the outgoing line to telephone 20 as it does normally, and in the case of a busy or ring-no-answer condition, the source platform 14 offers a feature by which the source platform 14 will call the caller back when the destination platform 18 is successful in contacting the called party. During the network camp-on service the network resources are dynamically deallocated when the service is requested and dynamically reallocated when the called party becomes available. When a busy or ring-no-answer occurs the system can also attempt to reach the called party via the Find-Me Service if it is active for the called party.

[0038] Call-Back-From-Call-Queue Service is similar to the Network-Camp-On feature discussed above, except it is used in the scenario where the called party is a service or help line that has the caller in a queue to be answered by the next available agent. Rather than having the calling party wait on hold, the destination platform 18 waits for a human being to answer the phone, and then tells the human being the customer is being contacted, and issues a network Call-Back, as discussed above, through source platform 14.

[0039] The Meet-Me (pager) service is a feature where the called party is a subscriber and has a pager number associated with the service. The destination platform 18 initiates a page, and adds a record to the subscriber database for the called party indicating there is a call holding for the subscriber on source platform 14. The called party upon receiving the page can initiate a call to the platform 18 (from anywhere), and the called party subscriber is told, upon entry, a call is holding, and allows the called party subscriber to accept the call. If the called party accepts the call, the calling party is taken off hold and the calling and called parties are connected together.

[0040] The Call-Forwarding feature is one where the called party subscriber record on the destination platform 18 has a different number for the subscriber. The different number is obtained by the source platform 14 from the destination platform 18 and used by the source platform 14 to initiate a call. The source platform 14 could either instruct the calling party the subscriber number has changed, or just use the different number without the caller's knowledge, based on the desires of the called party. Dynamic allocation and deallocation of network and platform resources occurs during call forwarding when the destination platform is released upon sending the source 14 the new telephone number and a new path through the network is allocated for the new telephone number.

[0041] Call-Interrupt is a feature where the called party allows a second caller to interrupt a conversation already occurring over the network 12 to telephone 20. When the second caller calls telephone 20, the destination platform 18 first checks (by looking in the subscriber database) to see if the called party subscriber is already on the line before placing the outgoing call session. If the called party subscriber is already on the line, the source platform 14 gives the caller the option to interrupt the call (if the called party subscriber allows this to occur) - or to notify the called party subscriber via a tone or series of tones played on telephone 20 indicating a call is waiting for the subscriber. The subscriber can then press a key to accept the new call. Call-Interrupt is also applicable when someone calls the calling party. By not sending voice packets to the calling party during an interrupted call, the resources of the packet-switched network are released for other uses.

[0042] In Business-Dialing a business has customer premise equipment (CPE) attached to the business PBX. The equipment provides access to the digital IP network 16. This access is provided by incorporating the functions of the present invention into the CPE. This allows the business to avoid the expensive per-minute access charges incurred whenever a call (local or long distance) is placed. This is also a savings to the digital IP network provider since it reduces the PSTN lines in use.

[0043] The Automatic Call Distribution (ACD) feature is one where a caller is put in a call queue for the called party subscriber (similar to the call waiting feature described above), except the queue is managed by the network to make sure calls are handled in the order in which they arrived. Virtual ACD is a similar feature, except the called party subscriber has a list of phone numbers used to accept incoming calls, and the network chooses the next available agent to process the call. Note the agents can be located anywhere on the network 12.

[0044] In Anywhere-Speed-Dial a calling party subscriber can have any number (typically up to 10) of speed dial numbers recognized by the platform 14.

[0050] Figure 3 depicts another embodiment which is typically used for long distance toll calls where the caller is charged for the individual calls. When the user places a toll call, such as by dialing "1" followed by the called party number, an end office 50 automatically transfers the call through the PSTN 12 to the platform 14, in the same way that end offices transfer toll calls to a preferred long distance provider today. At the same time, the calling party telephone number and the called party telephone number are transferred by conventional signalling, such as SS7 signalling, to the CU 30. The CU 30, in addition to assigning an APU 34 to handle the call and controlling switch 32, supplies the assigned APU 34 with the two telephone numbers. The APU 34 checks the subscriber database to confirm that the calling party telephone number is of a subscriber, and then looks up the destination platform address associated with the called party telephone number and places the call, the steps of which will be described in more detail later herein.

[0051] The internet telephony application of the APU 34 interacts with the calling party using a conventional interactive response type process to obtain the information needed to connect the call, such as calling party telephone number, calling party identification and called party telephone number. Once a connection has been established the APU 34 samples the incoming speech of the calling party (or called party) and compresses the speech using a compression procedure, such as the low bit rate procedure available from DSP Group as TrueSpeech/H.323 and which is typically used for the audio portion during video conferencing (see www.dspg.com for detailed information about this technology). The APU 34 forms speech segment packets having the IP address of the destination then sends the packets over the internal ethernet 38 to the NPU 36, which routes them over the network 16 to the destination. During the operation for establishing a link over the PSTN 12 to the called party telephone 20, the APU 34 performs a conventional out-dial process type process to "dial" the called party telephone and interact with the called party to establish the link. The APU 34 also performs call analysis, to be discussed in more detail later herein, and provides the results, such as indicating the called party telephone is "busy" or has hung-up, to the source platform 14 for communication to the calling party if appropriate. For speech data arriving over the Internet 16, the APU 34 performs the reverse operation to output speech signals to the telephone 10/20 through the switch 32 and PSTN 12.

[0052] Although not shown in Figure 2, the APU 34 preferably includes echo cancellation technology, such as available from Coherent Communications and which will be discussed in more detail later herein, and located on the PSTN "side" of the platform 14/18 to cancel echoes produced in the 2-to-4-wire hybrid circuits of the PSTN 12.

[0053] The platforms 14/18 also preferably perform a silence detection operation, which will also be discussed in more detail later herein, and do not send speech packets when the parties are not speaking, thereby conserving network resources. In this situation, the other side does not receive speech packets during a silent period and "plays" silence to the party.

[0054] Several different "connection" protocols can be used to establish a connection between two platforms over existing digital packet-switched-networks, including the internet protocol (IP), asynchronous transfer mode (ATM) packet switching protocol, frame relay, etc. The present invention can establish a connection and perform voice signal transfers using the Internet Protocol or other protocols. However, if the platforms are on the same ATM network, it is preferable that the Internet Protocol be used to establish the initial rendezvous, with the ATM protocol being used to establish an ATM connection, using ATM network IDs, between I/O ports of the platforms. Using an ATM connection provides improved delay characteristics as compared to an IP connection. The UDP protocol is preferably used whenever an ATM connection cannot be made.

[0055] During a period of UDP packet exchange between the platforms 14/18, such as when the calling and called parties are talking, each side sends each UDP packet with a monotonically increasing packet sequence number. The receiving side discards any packet arriving with a sequence number less than or equal to the packet sequence number of the packet currently playing (or which has completed playing).

[0056] The processes discussed below with respect to the present invention are described for convenience using flow charts depicting the operations as flowing from one operation to another. However, the processes are preferably implemented as interrupt driven processes. For example, during a conversation, the source platform 14 digitizes the speech of the subscriber and sends it to the destination platform 18 to be played to the caller, and when the caller hangs up, this is detected by the call monitoring operation of platform 14, which produces a hang-up interrupt detected by an interrupt manager that then executes the hang-up operations.

[0057] As previously mentioned, when a caller calls the platform 14, the caller selects the Internet telephony service and enters 100 a telephone number of the called party telephone, as depicted in Figure 4. When the caller has entered this destination telephone number, the source platform 14 APU (or CU) performs 102 a table look-up to obtain the name or address of the destination platform 18 servicing the area of the called party. The destination platform 18 is known by a specific machine name, such as "anp12.bostech.com." The source APU uses the Domain Name Service (DNS) of an Internet service provider (ISP) to translate the machine name of the platform 22 to its IP address. The source APU then connects 102 (issues a "TCP connect" using the TCP/IP) to the IP address returned by DNS, at a particular TCP port number (PN). The port number is

in the APU, to an energy threshold (including amplitude and time factors) below which the sample is considered to represent silence. If the sample is below this threshold no speech signal is transferred. In this way the network is not used for speech data transfer except when the parties are actually speaking. In this situation, the other side does not receive speech packets during a silent period and "plays" silence to the party. The silence detection operation preferably has a very short silence-to-energy detection threshold period (<5ms) to insure all the speech of the parties is captured and there is little or no speech clipping. The operation preferably has a relatively long energy-to-silence detection threshold period (>500ms). By meeting these criteria, computing resources are not wasted flipping back and forth between energy/silence. Several thresholds are preferably provided to allow distinguishing between different types of events, such as silence and a constant energy dial tone produced when a party hangs up, as well as to allow adjustments to occur when a telephone trunk is particularly noisy and would result in speech signals being transferred when the speaker is really silent or when a speaker speaks very quietly.

[0068] The segments of sound digitized and formed into a packet are preferably as small as possible. However, making a packet as small as 5 milliseconds (ms) of sound results in excessive addressing data overhead. A 20ms segment is an appropriate fixed and minimum sound segment size for a packet. A 20ms speech segment results in about 20 bytes of speech data which when combined with the 32 bytes of UDP-IP header data results in a packet with an overhead of over 50%. Preferably, the sound segment size of the packet is adjustable rather than fixed. The maximum delay determined during the initial call setup is used to adjust the size of the packets. The maximum delay is multiplied by 1.5 to obtain the segment size. If the network delay changes during the call, the sound segment size of the packet can be adjusted accordingly.

[0069] As the telephone call proceeds, the source and destination APUs monitor the energy of the trunks of the caller and called parties. When a hang up is detected 120/220 based on signaling from the PSTN, such as a ground start signal, the APU detecting the disconnect, say the destination APU, sends 222 a disconnect call packet to the other APU. The disconnect call packet is sent to particularly distinguish silence, when no data is being transferred, from an actual disconnect. The source APU terminates the billing event, stores the billing record and responds 122 with a disconnect acknowledgement packet. Both APUs stop 124/224 sampling, release their respective trunks (go on-hook) when the party hangs-up and indicate to the CU they and the trunks are available while the APUs terminate the virtual link between the platforms.

[0070] If the calling party does not hang-up after a period of time after the disconnect, the system can play a prompt to the calling party to start another call or some other service.

[0071] It is possible for the called party telephone to have an enhanced service, such as call screening. In such a situation, the call connect process previously discussed is not initiated until the destination APU has performed the call screening process to allow the called party telephone to be used to accept or reject the call from the calling party telephone. During such screening the caller receives a ringing signal initiated by the call analysis. If the called party refuses the call, the destination APU can immediately issue a disconnect to conserve network resources, with the source APU either continuing to play a ringing signal to the caller or providing a refusal message and prompting the caller to allow a message to be saved and delivered, such as described below.

[0072] The enhanced calling service called Message-Delivery typically is effective when the called party does not have a voice mail service or an answering machine. Message-Delivery, as depicted in Figure 6, starts with a call analysis operation that detects a busy (or ring-no-answer "RNA"). A busy packet is forwarded 240 to the source APU which plays 140 a busy tone to the caller. The platforms also proceeds through the call disconnect phase previously discussed to stop the sending of speech packets from the source platform 14 to the destination platform 18. At this time a prompt is overlaid on or multiplexed 142 with the busy tone (which busy tone is provided at reduced volume). The prompt asks the caller if the caller would like to leave a message for the destination. If the caller hangs up, or if the caller does not accept 144 the Message-Delivery Service, the trunk is released 146. If the caller accepts the service, a message is recorded 148. Once the entire message is recorded, including allowing the caller to change the message, etc. and indicate the message is to be sent, the source APU forwards 150 the entire message to the destination APU using the DPIPPN with a request to deliver the message. In doing this the source APU establishes a TCP/IP connection with the destination APU and conventionally transmits the message. The destination APU performs 242 a conventional outdial process to provide the message to the called party. When the called party receives the message, the destination APU sends 244 a completion message packet to the source APU. This packet can be used to bill for the message delivery or inform 152 the caller the message was received. Of course, if the message cannot be delivered within a predetermined period of time or the called party telephone has call screening and the called party does not accept the message, a failure message describing the problem can be provided to the caller.

[0073] During a ring-no-answer situation, after a ring indication has been received by the source platform 14 from the destination platform 18 and after a predetermined period of playing a ring signal (say 24 seconds) to the caller, a ring-no-answer prompt is played to the caller via the source APU overlaying the ring signal

ing the paged party to enter the page identification code, or if such a code is not used, by entering an account number and passcode. Once the paged party is identified, the destination APU plays a message indicating the call is being connected and simultaneously performs a connect operation to the source APU by sending 216 a call connect packet. The calling party and the paged party are connected and the call continues 116/218 as previously discussed. If the caller hangs-up before the paged party responds, the source APU informs the destination APU the caller has dropped the request. When the paged party calls the specific number, the destination APU responds to the paged party with the option of calling the caller back as occurs in the Call-Back service.

[0079] In the Call-Forwarding service, the source APU attempts to establish a call to the called party using a telephone number that, unknown to the source APU, has been forwarded to a different number. The destination APU, when it receives the service request, recognizes the number has changed using a look-up operation like in step 270 and responds to the source APU with a call forward indicator packet and the new telephone number. The source APU can inform the caller of the delay if desired. The source APU then starts a call connect process as if the caller had just entered this new number by performing the look-up 102, etc. necessary to establish a call to the new number.

[0080] In the Call-Interrupt/Call-Waiting service, when the subscriber calls the platform to obtain service, the platform records the subscriber's telephone is in use using the platform. When a destination platform 18 receives a service request prior to starting the outdial 208 for the telephone number, the destination APU checks the subscriber database in the CU to determine if the subscriber telephone number is currently in use. When the called party telephone is in use and the called party telephone has subscribed to the service, the destination APU sends a call progress indication to the source indicating there is a call in progress by the called party. The source APU plays a prompt to the caller indicating the called party telephone is in use and requesting the caller indicate whether the call should be interrupted, or whether the caller wants to hold or leave a message. The response of the user is relayed to the destination APU allowing the destination to disconnect if a message is being taken, play an interrupt tone and message to the called party if an interrupt is to occur or play a call-waiting tone if the user is waiting. The called party can accept the interrupt or switch to the waiting call by entering a code in the telephone. When the code is recognized by the destination APU monitoring the call, the destination APU, if the line is being using for another call, discontinues sending speech packets to the other location, discontinues playing speech packets from the other location and sends 216 a connect to the source and the call is connected as previously discussed. When the interrupting call is finished by the receipt of a hang-up packet from the other location or the entering of a switch back code by the called party, the exchange of speech packets with the other location continues. If the called party was involved in another operation, such as performing mailbox administration, the operation continues. It is also possible to bridge the two calls using a different code so all parties receive the speech packets creating a conference call.

[0081] In the Automatic-Call-Distribution (ACD) or virtual ACD service, the caller source address is placed in an ordered queue in the destination APU with the destination APU sending an indication of the entry into the queue to the source APU where a corresponding message is played to the caller. Traditionally, the caller remains in the line until an agent becomes available, however, it is possible to provide a call-back when the agent becomes available. The destination APU monitors call progress for multiple service personal or agent telephones and initiates a connect (216) between the first available agent and the caller at the top of the queue. The call progresses as has been previously discussed. In virtual ACD, the called party has agents located at different points in the packet-switched network 16, each with a different telephone number. When an agent signals to the agents APU that the agent is available for the next call, the agents APU accesses the queue and obtains the next caller from the list, including the IP address and port of the caller, and issues a connect directly to the caller. Once the connect occurs the agent and the caller talk. If the agent actually disconnects from the agents APU and frees up the APU port, the agent should be required to go through an authentication process to be revalidated as an agent. Because the connect is initiated by the agent's APU, the agent can be located anywhere on the packet-switched network.

[0082] In the Anywhere-Speed-Dial service, the source platform (typically on the CU) maintains a list of telephone numbers for the subscriber. When the subscriber calls the source APU, from any telephone, and enters the speed dial code and number indicator, the APU accesses the list and obtains the complete number and then starts the process with the look-up step 102 discussed with respect to Figure 4.

[0083] In the Call-Transfer service, when a transferring party[, say party 2,] (either a calling or called party) initiates a transfer operation, the APU, whether it is the source or destination APU, stops outputting (playing) the speech from the party, say party 1, to be transferred and stops sending speech signals to the party 1, essentially putting the party 1 on hold.

[0084] In the situation where the transferring party (party 2) does not want to speak with the party, say party 3, to whom the call is being transferred, what is called a blind transfer, the party 2 APU sends the telephone number and a transfer command to the transferred party 1's APU. The party 1 APU initiates a disconnect with party 2 and starts a new call by per-

packet-switched networks, such as an intranet. The invention can also be used with the emerging standards (incorporated herein by reference) being developed for the Internet, such as H.323, H.245 and T120 to allow the call to be a videophone call and allow easier call conferencing, and with emerging protocols (incorporated by reference herein), such as RSVP, which allows the reservation of resources for a desired level of service quality, and RTP and RTCP, which enhance the timeliness and synchronization of the packets.

[0093] The present invention has been described as requiring the caller to interact with the source platform to set up the call by inputing the called party telephone number. However, just as long distance telephone calls are routed to different carriers depending on the carrier selected by the caller, it is possible for a central office switch to route all long distance telephone calls from a particular telephone to a packet-switched-network calling platform and place the call over the packet-switched-network automatically and transparently.

[0094] The present invention has also been described with respect to conducting telephone calls between parties using conventional telephones. The invention is also applicable to a situation where one of the parties to a call is placing the call via a computer, such that a subscriber accessing the platform over the internet using a browser can "telephone" another individual conversing using a telephone.

Claims

1. A telephone system, comprising: first and second telephones;

   a packet-switched-network; and
   first and second interface conversion units coupled to said packet-switched-network and respectively to said first and second telephones, said units exchanging speech data packets over said network and converting packets into speech signals and speech signals into packets and dynamically allocating resources of said packet-switched-network as needed to complete a telephone call transaction.

2. A system as recited in claim 1, wherein the resources of said first and second interface conversion units are dynamically allocated to complete said transaction.

3. A system as recited in claim 1 or 2, wherein said first and second interface conversion units exchange network addresses used for exchanging the speech data packets.

4. A system as recited in any of the preceding claims, wherein said first interface conversion unit determines a network address of said second interface conversion unit using a telephone number.

5. A system as recited in claim 4, wherein said first interface conversion unit transmits a packet switched message to a said second interface conversion unit including a network address of said first interface conversion unit.

6. A system as recited in claim 5, wherein said second interface conversion unit replies to said request with an asynchronous transfer mode network address and said first and second interface conversion units establish a virtual connection over said packet-switched-network.

7. A telephone system, comprising:

   first and second telephones;
   a packet-switched-network; and
   first and second interface conversion units coupled to said packet-switched-network and respectively to said first and second telephones, said units exchanging speech data packets over said network, converting packets into speech signals and speech signals into packets, and compressing the speech signals prior to conversion into speech data packets.

8. A telephone system, comprising:

   first and second telephones;
   a packet-switched-network; and
   first and second interface conversion units coupled to said packet-switched-network and respectively to said first and second telephones, said units exchanging speech data packets over said network and converting packets into speech signals and speech signals into packets and providing enhanced calling services including one of: message-delivery, call-back, call-screening, network-camp-on, call-back-from-queue, find-me, meet-me-pager, call-forwarding, call-interrupt, call-waiting, business-dialing, automatic-call-distribution, virtual-automatic-call-distribution, speed-dial, call-transfer, conference-calling and network-administration.

9. A telephone system, comprising:

   a telephone;
   a packet-switched-network; and
   an interface conversion unit coupled to said packet-switched-network and to said telephone, said unit converting speech signals from said telephone into speech data packets, transmitting the speech data packets over said

15. A method as recited in claim 13 or 14, further comprising exchanging packet-switched-network addresses associated with the caller and called telephones.

16. A method as recited in any of claims 13 to 15, further comprising establishing a virtual circuit over the packet-switched-network.

17. A telephone call method, comprising: converting speech signals from a calling telephone into calling party speech data packets when the speech signals are above a speech signal threshold; transmitting the calling party speech data packets to a called party destination using a packet-switched-network; and converting the calling party speech data packets into called party speech signals.

18. A telephone call method, comprising: converting speech signals from a calling telephone into calling party speech data packets; transmitting the calling party speech data packets to a called party destination using a packet-switched-network;

converting the calling party speech data packets into called party speech signals; and
determining a network delay and adjusting a sound segment size of the data packets responsive to the delay.

19. A method as recited in claim 18, wherein the size comprises 1.5 times the delay.

20. A telephone system, comprising:

first and second telephones;
a packet-switched-network; and
first and second end points coupled to said packet-switched-network and respectively to said first and second telephones, said end points managing a telephone call between said first and second telephones over said network.

21. A telephone system as recited in claim 20, wherein said end points comprise means for providing enhanced calling services including one of: message-delivery, call-back, call-screening, network-camp-on, call-back-from-queue, find-me, meet-me-pager, call-forwarding, call-interrupt, call-waiting, business-dialing, automatic-call-distribution, virtual-automatic-call-distribution, speed-dial, call-transfer, conference-calling and network-administration.

22. A telephone call method, comprising:

managing a telephone call over a packet-switched network between source and destination end points.

23. A method as recited in claim 22, wherein said managing provides enhanced calling services including one of: message-delivery, call-back, call-screening, network-camp-on, call-back-from-queue, find-me, meet-me-pager, call-forwarding, call-interrupt, call-waiting, business-dialing, automatic-call-distribution, virtual-automatic-call-distribution, speed-dial, call-transfer, conference-calling and network-administration.

24. A method as recited in claim 22 or 23, wherein said managing comprises:

initiating a packet-switched packet communication between the end points; and
performing packet communication between the end points to establish a call between the end points.

25. A method as recited in any of claims 22 to 24, wherein said managing comprises:

initiating a packet-switched packet communication between the end points; and
performing packet communication between the end points to establish an enhanced telephone service between the end points.

26. A method as recited in any of claims 22 to 25, wherein said managing comprises dynamically allocating network resources as needed to complete a telephone call transaction.

27. A method as recited in any of claims 22 to 26, wherein said managing comprises dynamically allocating end point resources as needed to complete a telephone call transaction.

28. A telephone system, comprising:

first and second telephones;
a network; and
first and second interface conversion units coupled to said network and respectively to said first and second telephones, said units exchanging speech data packets over said network, converting packets into speech signals and speech signals into packets, and dynamically allocating resources of said network and said units as needed to complete a telephone call transaction.

FIG. 2

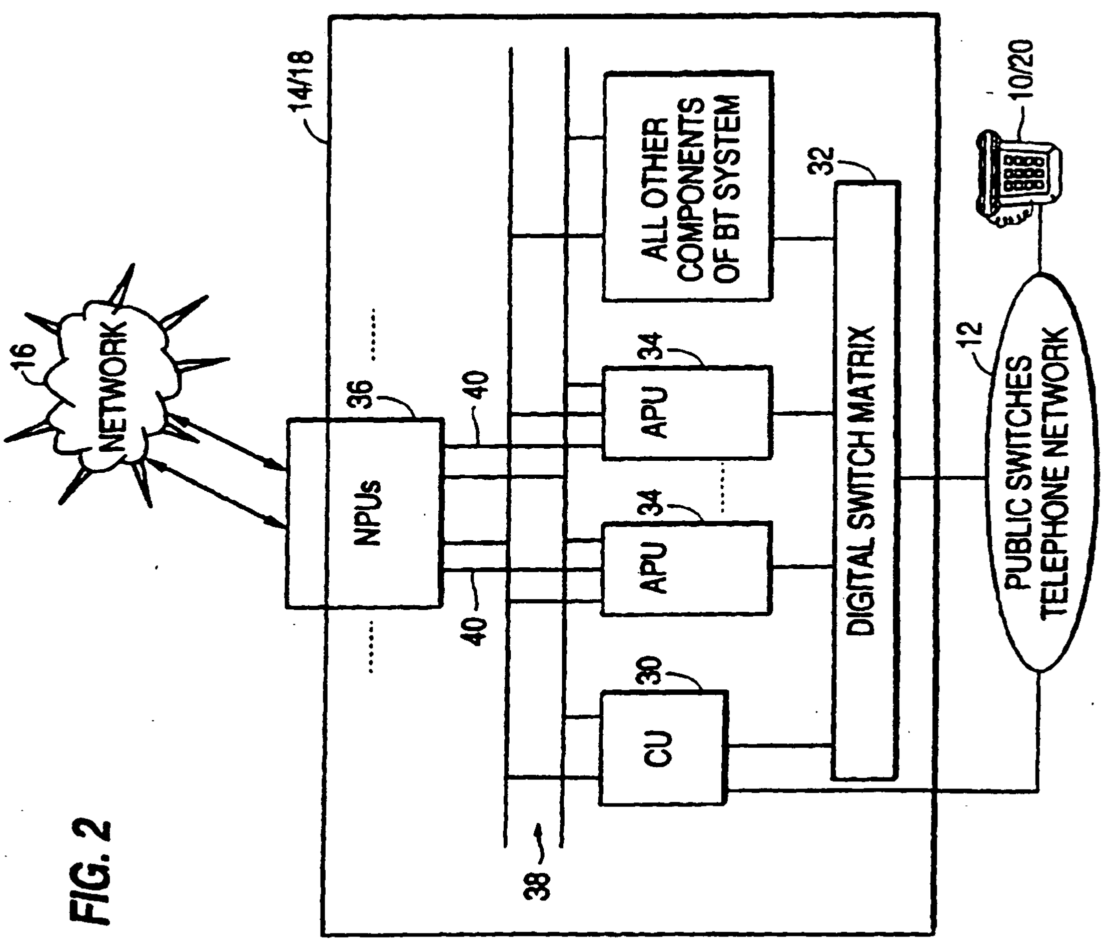

16
NETWORK
14/18
36
NPUs
40
40
34
APU
34
APU
30
CU
ALL OTHER COMPONENTS OF BT SYSTEM
32
DIGITAL SWITCH MATRIX
38
12
PUBLIC SWITCHES TELEPHONE NETWORK
10/20

FIG. 4

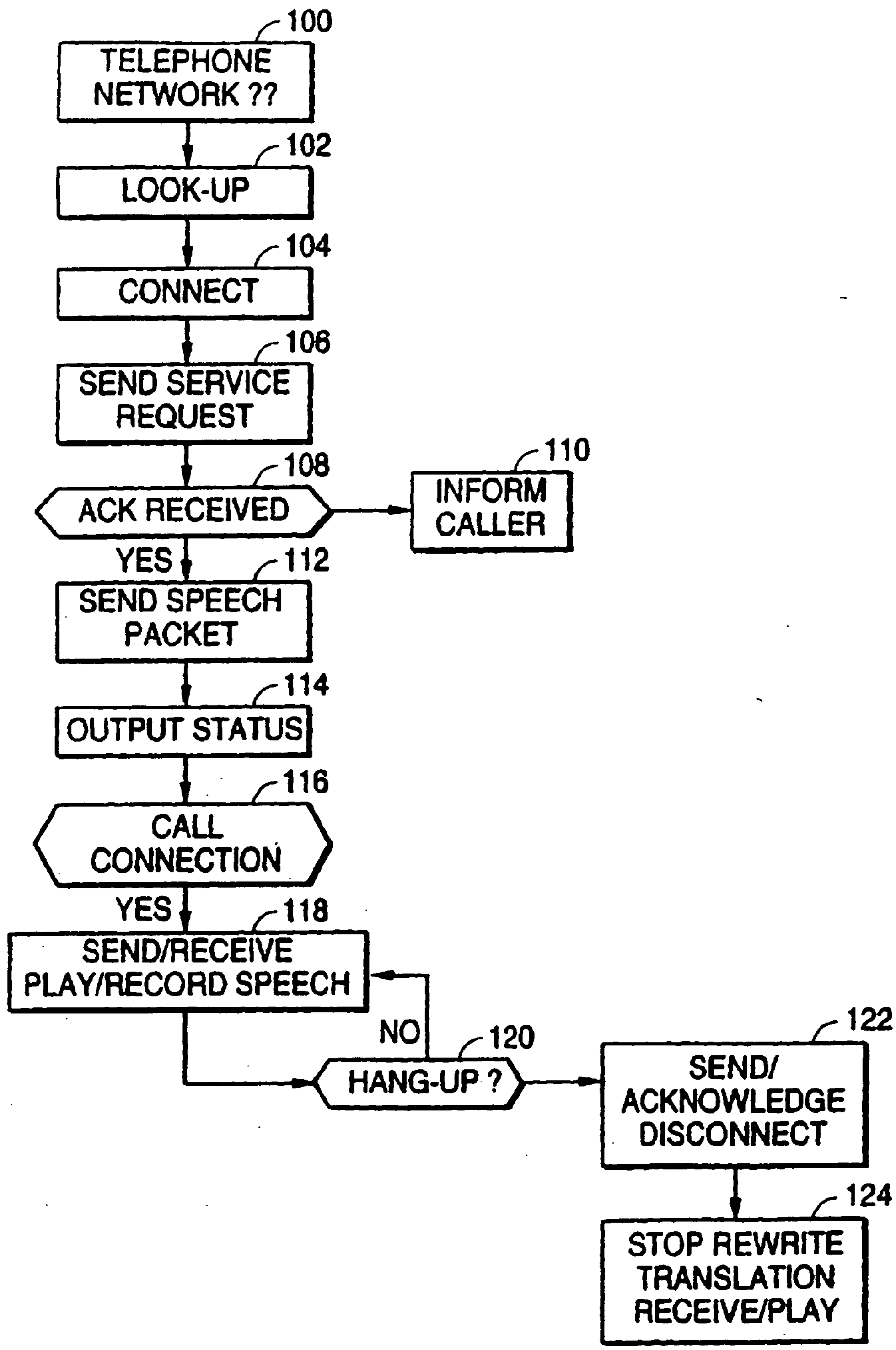

100
TELEPHONE NETWORK ??
102
LOOK-UP
104
CONNECT
106
SEND SERVICE REQUEST
108
ACK RECEIVED
110
INFORM CALLER
YES
112
SEND SPEECH PACKET
114
OUTPUT STATUS
116
CALL CONNECTION
YES
118
SEND/RECEIVE PLAY/RECORD SPEECH
NO
120
HANG-UP ?
122
SEND/ ACKNOWLEDGE DISCONNECT
124
STOP REWRITE TRANSLATION RECEIVE/PLAY

FIG. 6

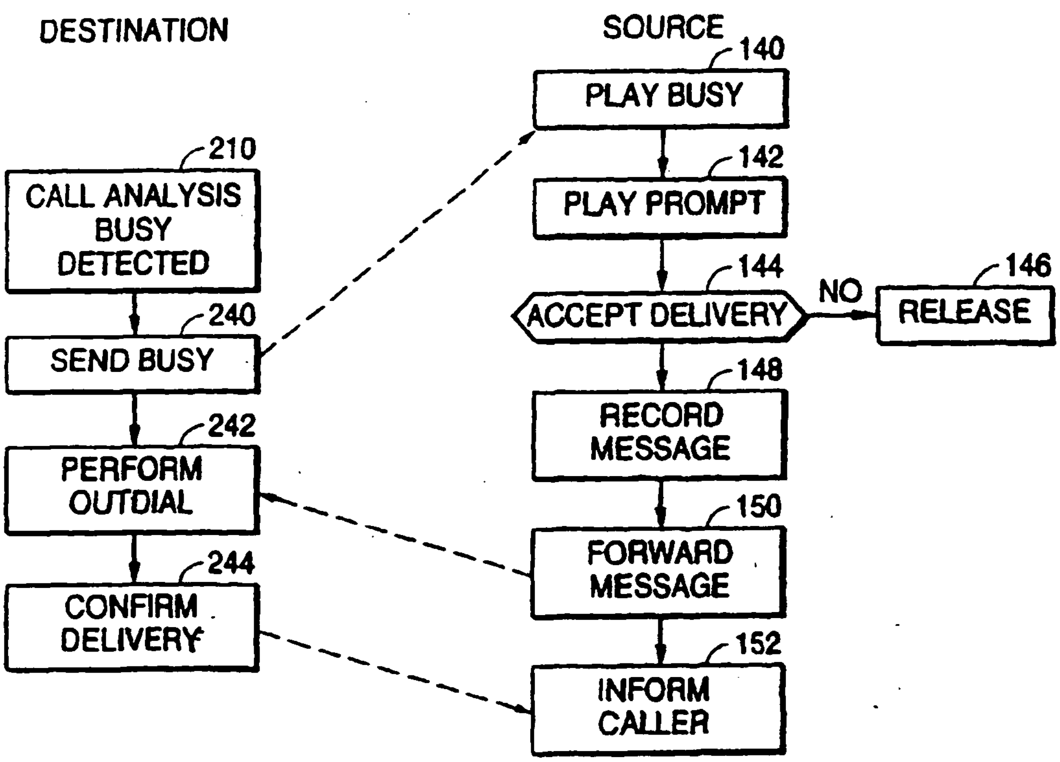
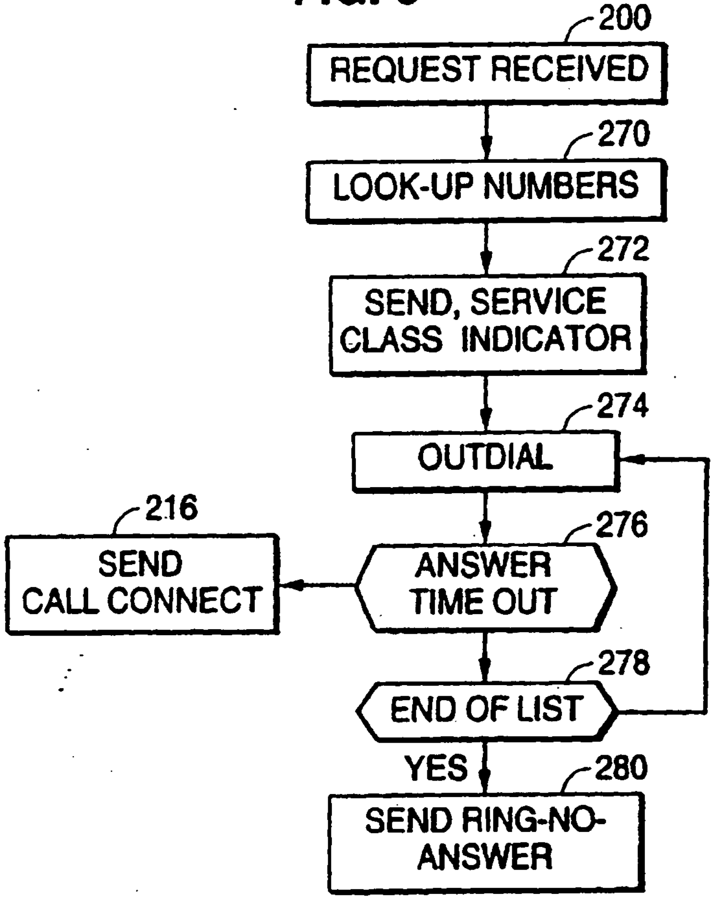

DESTINATION
SOURCE
140
PLAY BUSY
210
CALL ANALYSIS BUSY DETECTED
142
PLAY PROMPT
144
146
NO
ACCEPT DELIVERY
RELEASE
240
SEND BUSY
148
RECORD MESSAGE
242
PERFORM OUTDIAL
150
FORWARD MESSAGE
244
CONFIRM DELIVERY
152
INFORM CALLER

FIG. 8


200
REQUEST RECEIVED
270
LOOK-UP NUMBERS
272
SEND, SERVICE CLASS INDICATOR
274
OUTDIAL
276
ANSWER TIME OUT
216
SEND CALL CONNECT
278
END OF LIST
YES
280
SEND RING-NO-ANSWER

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Date; 29/06/2000